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(54) Audio Signal Processing Apparatus with Optimization Process

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ABSTRACT OF THE DISCLOSURE

An audio signal processing apparatus performs the optimization process of an audio signal reproduced from a medium such as a compact disk. Each of low-pass and high-pass filters effects a band division on the audio signal. A peak analyzer detects the peak value of a low audio-range signal and outputs a control signal corresponding to the detected peak value to a voltage-controlled amplifier. The voltage-controlled amplifier adjusts the level of the low audio-range signal in accordance with the control signal. A mixture and distribution circuit combines the output of the voltage-controlled amplifier and the output of the high-pass filter and distributes the composite signal to corresponding power amplifiers for respectively driving speakers. Thus, a process suited to each of various music sources can be achieved by controlling the level of a low audio-range component based on the level of the low audio-range signal.

AUDIO SIGNAL PROCESSING APPARATUS WITH OPTIMIZATION PROCESS

BACKGROUND OF THE INVENTION

Field of the Invention

The present invention relates to an audio signal processing apparatus which is mounted on a vehicle or the like and effects the optimization process on a reproduced audio signal.

Description of the Related Art

Various tone-quality controlling functions have heretofore been proposed for an audio reproducing or playback device. The audio playback device now tends toward improvement in the tone quality as the whole audio system while these functions are being associated with the convenience for the use of a user. In an audio system mounted on a vehicle in particular, a function for supplementing the deficiency of intensity of volume in a low audio range under the audibility attributed to or affected by noise on travelling, has been realized. For example, a low audio-range boosting function, a graphic equalizer function for correcting an acoustic characteristic in a vehicle room and an auto loudness function for varying a frequency response in cooperation with volume control taken to correct a loudness

characteristic on the mentality of hearing have been put to practical use.

To enable the prior art to be described with the aid of diagrams the figures of the drawings will first be listed.

FIG. 1 is a schematic view showing the structure of a vehicle audio system including a conventional audio signal processing apparatus;

FIG. 2 is a graph showing a frequency spectrum for a hard-rock music signal;

FIG. 3 is a waveform chart for describing the waveform of the hard-rock music signal with respect to time;

FIG. 4 is a block diagram showing a conventional improved audio signal processing apparatus;

FIG. 5 is a block diagram illustrating the structure of a conventional waveform compression circuit;

FIG. 6 is a block diagram depicting the structure of another conventional waveform compression circuit;

FIG. 7 is a view for describing frequency vs. sound level curves;

FIG. 8 is a view for explaining a spectrum pattern for a classic or pops music signal;

FIG. 9 is a view for describing a spectrum pattern for a rock music signal;

FIG. 10 is a schematic view illustrating the structure of a vehicle audio system including an audio signal processing apparatus according to one embodiment of the present invention;

FIG. 11 is a circuit diagram showing one example of the configuration of a peak analyzer;

FIG. 12 is a circuit diagram depicting one example of the configuration of a voltage-controlled amplifier;

FIG. 13 is a waveform chart for explaining the manner of changes in waveforms of audio signals;

FIG. 14 is a block diagram showing an audio signal processing apparatus according to another embodiment of the present invention;

FIG. 15 is a block diagram illustrating an audio signal processing apparatus according to a further embodiment of the present invention;

FIG. 16 is a circuit diagram showing one example of the configuration of a low audio-range boosting circuit;

FIG. 17 is a block diagram illustrating an audio signal processing apparatus according to a still further embodiment of the present invention;

FIG. 18 is a block diagram showing an audio signal processing apparatus using a digital circuit, according to a still further embodiment of the present invention;

FIG. 19 is a circuit diagram showing one example of the configuration of a quadrative mirror filter used for the division of a band;

FIG. 20 is a circuit diagram illustrating one

example of the configuration of a quadrative mirror filter used for the synthesis of signals;

FIG. 21 is a circuit diagram showing one example of the configuration of a IIR filter;

FIG. 22 is a block diagram depicting an audio signal processing apparatus using a digital circuit, according to a still further embodiment of the present invention;

FIG. 23 is a block diagram showing an audio signal processing apparatus using a DSP, according to a still further embodiment of the present invention;

FIG. 24 is a block diagram illustrating an audio signal processing apparatus using a DSP, according to a still further embodiment of the present invention;

FIG. 25 is a flowchart for describing a sequential process for analyzing and compressing a waveform;

FIG. 26 is a waveform chart for describing the manner of processing for the waveform analysis and compression;

FIG. 27 is a flowchart for explaining one example of a waveform analyzing process;

FIG. 28 is a waveform chart for explaining one example of the result of the waveform analyzing process;

FIG. 29 is a waveform chart for explaining another example of the result of the waveform analyzing process;

FIG. 30 is a flowchart for describing another example of a waveform analyzing process; and

FIG. 31 is a waveform chart for describing one example of the result of the waveform analyzing process.

FIG. 1 is a schematic view showing the structure of a conventional audio system suitable for use in a vehicle. Here, a four-speaker on-board system is shown. In the same drawing, there are shown an audio playback device 1 such as a CD player, a cassette tape player or the like, an audio signal processing circuit 2 having the low audio-range boosting function, the loudness control function, the graphic equalizer function, etc., for effecting signal processing on stereo signals supplied from the audio playback device 1, speakers 21 through 24 disposed within a vehicle room 200, and power amplifiers 31 through 34 for amplifying signals supplied from the audio signal processing circuit 2 and supplying the amplified signals to their corresponding speakers.

The operation of the audio system will next be described. The audio playback device 1 reproduces the stereo signals from an audio recording medium and outputs the same to the audio signal processing circuit 2. The audio signal processing circuit 2 effects signal processing on the stereo signals in accordance with the aforementioned functions and distributes the processed signals to their corresponding power amplifiers 31

through 34. The power amplifiers 31 through 34 effect power amplification on the input signals and output the amplified signals to their corresponding speakers 21 through 24. The speakers 21 through 24 output or radiate acoustic power based on the input signals. Thus, an audible sound is reproduced with the vehicle room 200 as a sound field.

In the vehicle audio system, however, restrictions are imposed on the size, weight and mounting position of each of the speakers 21 through 24. Thus, a limit is placed upon the ability of reproduction of a heavy low-pitched sound due to such restrictions. When signals exceeding the limit (permissible capability) are applied to the speakers 21 through 24, the sound is distorted so that a discomfortable sound field is offered. Therefore, the audio signal processing circuit 2 specifies the maximum output voltage and controls in such a manner that voltages output from the power amplifiers 31 through 34 do not exceed the allowable inputs of the speakers 21 through 24.

Alternatively, there has been adopted a means for avoiding the application of excessive inputs to their corresponding speakers 21 through 24 using a characteristic for clipping a power source voltage with

the power amplifiers 31 through 34. In this case, however, the distortion developed in the speakers 21 through 24 is reduced but large distortion appears at the power amplifiers 31 through 34. Therefore, the quality of audible tones is deteriorated.

In the conventional vehicle audio system, a means for bringing out the capability of reproduction, which is well balanced as a whole, has been made based on the above-described construction. When, however, signals, which are wide in dynamic range, such as signals produced from a digital audio medium such as a typical compact disk, are reproduced, a problem arises in terms of the capability of reproduction. Further, when it is desired to reproduce music of various genres from classical music to hard-rock music, a problem is aggravated.

As is understood from a frequency spectrum shown in FIG. 2, low audio-range frequency components of from 40Hz to 100Hz are excessively large in amplitude in the case of hard-rock music signal, for example. Even in the case of a waveform shown in FIG. 3 on a time basis, the height or amplitude of a waveform corresponding to low audio-range components extends or varies upward and downward. When a music signal having such a waveform is reproduced by a system using

a conventional conception, distortion appears at power amplifiers and speakers due to the existence of the super low audio-range components, with the result that a distorted reproduced sound is produced.

In order to prevent the distortion from occurring, the design of circuits for reducing the entire amplification factor is normally carried out. However, when classic and pops music signals are reproduced under a system based on such a design, a recording level is low. Thus, non-powerful and unsatisfactory sound is reproduced even when a volume control level is set to the maximum.

In order to cope with such a problem, there has been proposed an audio signal processing apparatus for controlling the level of a waveform whose level is excessively large by effecting waveform compression upon amplification of an audio signal with a nonlinear amplifying means thereby to prevent an excessive input from being applied to each of speakers. As a typical example of this type of audio signal processing apparatus, there is known one in which the power source voltage clipping characteristic of each power amplifier referred to above is used as a nonlinear characteristic. However, according to the audio signal processing

apparatus using the power source voltage clipping characteristic, the amount of distortion varies according to the degree of nonlinearity but the quality of a tone is deteriorated at each power amplifier as an alternative to the control of distortion which appears at each speaker.

Thus, a system free of the usage of the nonlinear characteristic of each power amplifier has been proposed in which as shown in FIG. 4, a waveform compression circuit 7A serving as a nonlinear amplifying means is provided between an audio signal processing circuit 2 and power amplifiers 31 and 32 and each waveform is compressed by the waveform compression circuit 10 upon amplification of an audio signal thereby to control the level of a waveform whose level is excessively large. Here, a two-speaker system will be described by way of example.

FIG. 5 shows a specific configuration of the waveform compression circuit 7A. In this type of configuration, a signal input from the audio signal processing circuit 2 is first amplified by an amplifier 11 and then detected by a detector 12. Thereafter, the signal detected by the detector 12 is smoothed with a predetermined time constant by a time constant circuit 13. Thus, an average detected level is output from the

time constant circuit 13. A voltage-controlled amplifier 14 varies an amplification factor according to the detected level and amplifies the input signal based on the amplification factor so as to be supplied to each of the power amplifiers 31 and 32 as an output signal.

As another specific configuration of the waveform compression circuit 7A, there is known one disclosed in Japanese Utility Model Application Laid-Open No. 63-35311 as illustrated in FIG. 6. In this configuration, an output signal is first amplified by an amplifier 11 and then detected by a detector 12. Thereafter, the detected signal supplied from the detector 12 is smoothed with a predetermined time constant by a time constant circuit 13 from which an average detected level is produced. Further, a voltage-controlled amplifier 14 varies an amplification factor according to the detected level and amplifies the input signal based on that amplification factor so as to be supplied to a low-pass filter circuit 17. The low-pass filter circuit 17 allows the passage of low audio-range frequency components of the output supplied from the voltage-controlled amplifier 14. In addition, a high-pass filter circuit 15 allows the passage of high audio-range frequency components of the signal input from an audio signal processing circuit 2. A mixing circuit 16

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combines the output of the low-pass filter circuit 17 and the output of the high-pass filter circuit 15 and supplies the result of combination to each of power amplifiers 31 and 32 as the output signal.

The waveform compression circuit 10 shown in FIG. 5 detects the level of the input signal, while the waveform compression circuit 10 shown in FIG. 6 detects output signal to be processed. However, both circuits make use of an audio signal on the whole band as an object for execution of the level detection for the signal processing. Further, the output produced as the detected level corresponds to the average level which has been smoothed with the predetermined time constant.

Since the conventional audio signal processing apparatus is constructed as described above, there is often a situation in which the waveform compression is made in response to the level of a high audio-range portion unnecessary to originally effect the waveform compression due to the fact that the signal on the whole band is defined as the object for the level detection. Thus, a problem arises that the quality of tones is deteriorated and a sound escape or leakage phenomenon takes place that the sound is lowered beyond need.

Further, a problem arises that since the waveform compression is carried out according to the

average level, nonlinear distortion at the rising maximum peak value cannot be avoided by the waveform compression during damped oscillation which often appears at the waveform of a rock music signal shown in FIG. 2.

As an alternative to the above art, other prior art has been disclosed in U.S. Pat. No. 4,398,061, which shows a system for detecting the peak value of the waveform of an audio signal, which appears between adjacent zero cross-points on the waveform and compressing each waveform between the zero cross-points according to the peak value. According to such a system, distortion developed due to a nonlinear characteristic of an audio signal playback system and an excessive increase in peak value can be sufficiently reduced. However, since a signal on the whole band is defined as an object for the detection of its peak, the compression is effected even on projected peak-value components in intermediate or middle high audio ranges. As a result, the sound leakage phenomenon sometimes appears in the same manner as described above and hence a sense of incompatibility cannot be avoided in terms of audibility.

According to the audio signal processing apparatus employed in the conventional vehicle audio

system, as has been described above, a suitable sound to be heard cannot be obtained from a wide range of audio sources. The largest factor is that a signal processing system has not yet been designed which makes good use of knowledge on psychological acoustics representing what knowledge about the mode of a music signal and the sense of distortion with respect to a reproduced sound are.

SUMMARY OF THE INVENTION

The present invention has been made to solve the foregoing problems. Therefore, the object of the present invention is to provide an audio signal processing apparatus for providing a reproduced sound suitable for a wide range of music, using the fact that a difference between respective music genres appears in low audio-range components of a music signal.

Here, knowledge on acoustic psychology, which forms the basis of the present invention, will be described. A human sense of loudness depends on both the frequency and level of a sound. The results of investigation with respect to the loudness sense have been reported in various form. However, the human sense of loudness is basically represented in the form of characteristics shown in FIG. 7. In the same drawing, the axis of ordinate represents a sound level. This

drawing is obtained by plotting respective sound levels at other frequencies each audible at the loudness equal to a sound level at a predetermined frequency (1kHz, for example).

According to the characteristics shown in FIG. 7, the sensitivity of middle and low audio ranges is substantially flat in the case of a loudness represented in the form of an audible sound pressure exceeding 100dB. In the case of a low loudness, the sensitivity of a low audio range is greatly dulled or reduced. It is considered that the hearing level at the time that music is being appreciated in a silent room, is normally of about 80dB on the average. When the hearing level is in the vicinity of 60dB to 100dB, the sensitivity at a frequency of 100Hz or less is dulled by 5dB to 20dB over the sensitivity at a frequency in the vicinity of 1kHz. Since noise components of 100Hz or less are high in level or amplitude in the running vehicle room 200, the sensitivity of the loudness in the low audio range becomes progressively worse.

It is thus necessary to effect, during the running of a vehicle, a corresponding sensitivity correction like low audio-range boosting on classic and pops music signals whose low audio-range frequency components are not so high in level as represented by a

spectrum pattern of FIG. 8. However, the low audio-range boosting is not necessarily effected on a rock music signal whose low audio-range frequency components are high in level as represented by a spectrum pattern of FIG. 9 where the sound is heard in loud volume.

According to the knowledge on the acoustic psychology referred to above, it is understood that it is necessary to correct the sensitivity of the loudness according to the music genres and the hearing level. That is, the present invention has been made based on the aforementioned knowledge.

An audio signal processing apparatus according to a first aspect of this invention comprises frequency band-dividing filter means for by-band dividing an audio signal into a low audio-range signal and a high audio-range signal, waveform analyzing means for analyzing the peak value of the low audio-range signal divided by the band-dividing filter means, waveform compressing means for compressing the level of a waveform of the low audio-range signal according to the result of analysis by the waveform analyzing means, and mixing means for blending the low audio-range signal compressed by the waveform compressing means and the high audio-range signal supplied from the band-dividing filter means. The waveform analyzing means analyzes the peak value of

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only the low audio-range signal in the audio signal. Further, the waveform compressing means compresses the level of the low audio-range signal according to the peak value of the low audio-range signal in the audio signal.

An audio signal processing apparatus according to a second aspect of this invention comprises frequency band-dividing filter means for by-band dividing an audio signal into a low audio-range signal and a high audio-range signal, waveform analyzing means for analyzing the peak value of the low audio-range signal divided by the band-dividing filter means, delaying means for supplying a delay of a time interval including a time interval required for the processing of analysis by the waveform analyzing means to the low audio-range signal and the high audio-range signal supplied from the band-dividing filter means, waveform compressing means for compressing the level of a waveform of the delayed low audio-range signal according to the result of analysis by the waveform analyzing means, and mixing means for combining the low audio-range signal compressed by the waveform compressing means and the high audio-range signal delayed by the delaying means. The delaying means delays the signals supplied from the band-dividing filter means in such a manner that the result of

analysis by the waveform analyzing means is reflected in a waveform to be analyzed.

An audio signal processing apparatus according to a third aspect of this invention is constructed by additionally disposing volume controlling means for controlling the level of the audio signal and compression-rate controlling means for reflecting the rate of control by the volume controlling means in the result of analysis by the waveform analyzing means in the audio signal processing apparatus according to each of the first and second aspects. The compression-rate controlling means adds the rate of control by the volume controlling means to the result of analysis by the waveform analyzing means thereby to allow the waveform compressing means to effectively compress each waveform based on the result of addition.

An audio signal processing apparatus according to a fourth aspect of this invention is set in such a manner that the frequency band-dividing filter means, the waveform analyzing means and the waveform compressing means are respectively made up of digital processing circuits. Thus, the band-dividing filter means comprises a digital filter and more precisely effects the band division of the audio signal.

An audio signal processing apparatus according

to a fifth aspect of this invention is set in such a manner that the band-dividing filter means is comprised of a digital mirror filter. Thus, the band division of the audio signal is achieved more strictly.

An audio signal processing apparatus according to a sixth aspect of this invention is set in such a manner that the frequency band-dividing filter means, the waveform analyzing means and the waveform compressing means are constructed in the form of a digital signal processing integrated circuit. Thus, circuit configurations of the entire audio signal processing apparatus are simplified.

An audio signal processing apparatus according to a seventh aspect of this invention is constructed in such a manner that the amount of delay by the delaying means is set to a range of 5ms to 100ms. Thus, the delaying means supplies the amount of delay corresponding to a characteristic of an audio signal to an input signal thereby to control the unnaturalness of a reproduced sound.

An audio signal processing apparatus according to an eighth aspect of this invention is constructed in such a manner that the waveform analyzing means effects waveform analysis with a time interval between adjacent zero cross-points on a waveform as a unit. Since the

time interval between the zero cross-points is defined as one analysis period, the waveform analyzing means is activated so as to prevent waveform distortion from being produced between a certain analysis period and the next analysis period.

An audio signal processing apparatus according to a ninth aspect of this invention is constructed in such a manner that the waveform analyzing means effects waveform analysis with each waveform within a fixed time interval as a unit. The waveform analyzing means can simplify circuits with the analysis period as fixed.

The above and other objects, features and advantages of the present invention will become apparent from the following description and the appended claims, taken in conjunction with the accompanying drawings in which preferred embodiments of the present invention are shown by way of illustrative example.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENTS

[First Embodiment]

FIG. 10 is a view for explaining the structure of a vehicle audio system including an audio signal processing apparatus according to a first embodiment of the present invention. In the same drawing, there are shown an audio reproducing or playback device 1 such as a CD player, a cassette tape player or the like, an audio signal processing circuit 2 having a low audio-range boosting function, a loudness control function, a graphic equalizer function, etc., for effecting signal processing on stereo signals supplied

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from the audio playback device 1, a volume controller 3, a low-pass filter 4 for allowing low audio-range frequency components of audio signals subjected to the volume control by the volume controller 3 to pass therethrough, a high-pass filter 5 for allowing high audio-range

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frequency components of the audio signals subjected to the volume control by the volume controller 3 to pass therethrough, a pulse peak analyzer 6 for analyzing the peak value of a waveform of each audio signal which falls within a low audio range, a voltage-controlled amplifier (VCA) 7 for amplifying a signal output from the low-pass filter 4 in response to the output of the peak analyzer 6, a mixture and distribution circuit 8 for mixing the output of the voltage-controlled amplifier 7 and the output of the high-pass filter 5 and distributing output signals to their corresponding power amplifiers 31 through 33, and speakers 21 through 24 mounted in a vehicle room 200.

Incidentally, the low-pass filter 4 and the high-pass filter 5 show a practical example of a band-dividing filter means and the peak analyzer 6 shows a practical example of a waveform analyzing means. Further, the voltage-controlled amplifier 7 illustrates a practical example of a waveform compressing means and the mixture and distribution circuit 8 shows a practical example of a mixing means.

The operation of the audio signal processing apparatus shown in FIG. 10 will next be described below. The audio playback device 1 reproduces stereo signals from an audio recording medium and outputs the same to

the audio signal processing circuit 2. The audio signal processing circuit 2 effects signal processing on the input stereo signals in accordance with a low audio-range or bass boosting function, a graphic equalizer function for correcting an acoustic characteristic in a vehicle room and an auto-loudness function for varying a frequency characteristic in cooperation with the volume control effected with a view toward correcting a loudness characteristic on the mentality of hearing. Thereafter, the audio signal processing circuit 2 outputs the so-processed signals to the volume controller 3. The volume controller 3 effects volume control processing on the input signals in accordance with a volume setting operation by a user.

The output of the volume controller 3 is supplied to the low-pass filter 4 and the high-pass filter 5. The low-pass filter 4 allows the passage of low audio-range frequency components of the signals input from the volume controller 3. The input signals are normally of audio signals each having a frequency range of 20Hz through 20kHz. Thus, the cut-off frequency of the low-pass filter 4 is set to one of 100Hz through 200Hz. The high-pass filter 5 has a characteristic held in a supplementary relationship to a characteristic of the low-pass filter 4 and allows the

passage of high audio-range frequency components of the signals input from the volume controller 3. Incidentally, each of the low-pass filter 4 and the high-pass filter 5 may be used as a quadratic active filter. It is also preferable that the complementarity in cut-off characteristic between the low-pass filter 4 and the high-pass filter 5 is ensured.

The signals output from the low-pass filter 4 are sent to the peak analyzer 6 and the voltage-controlled amplifier 7. The peak analyzer 6 detects the peaks of the output signals of the low-pass filter 4 at a predetermined time interval. Further, the peak analyzer 6 compares the peak levels of the input signals with a predetermined level and outputs a control voltage corresponding to the degree or rate of waveform compression depending on the result of comparison to the voltage-controlled amplifier 7. The voltage-controlled amplifier 7 then compresses level of the signal output from the low-pass filter 4 in accordance with the control voltage.

FIG. 11 is a block diagram showing one example of a specific circuit configuration of the peak analyzer 6. However, only one channel of stereo two channels is shown herein. In the same drawing, reference numeral 40 indicates a peak holding circuit comprised of

operational amplifiers 41 and 42, a diode 43, a peak holding capacitor 44 and a resetting analog switch 45.

The signals produced from the low-pass filter 4 are supplied to a non-inversion input terminal of the operational amplifier 41. The output of the operational amplifier 41 is supplied to the peak holding capacitor 44 and a non-inversion input terminal of the operational amplifier 42 through the diode 43. The output of the operational amplifier 42 is produced as a detected output indicative of peak value and fed back to inversion input terminals of the operational amplifiers 41 and 42.

Designated at numeral 50 is a reset pulse generator circuit which comprises diodes 52 and 53, resistive elements 54 through 57 and a capacitor 58. A series resistor comprised of the two resistor elements 56 and 57 is electrically connected between both polarities of a power source E1. The point at which the resistor elements 56 and 57 are connected to each other, is electrically connected to an inversion input terminal of the operational amplifier 51. The output of the operational amplifier 51 is produced as the output of the reset pulse generator circuit 50 and fed back to a non-inversion input terminal of the operational amplifier 51 through resistor 54. A series body

comprised of the resistor element 55 and the diode 52 is electrically connected between the output terminal and non-inversion input terminal of the operational amplifier 51. The diode 52 is electrically connected in reverse parallel with the diode 53. The non-inversion input terminal of the operational amplifier 51 is grounded through the capacitor 58.

An astable multivibrator is made up of the operational amplifier 51, the diodes 52 and 53, the capacitor 58 and the resistor elements 54 through 57 all of which have been connected to one another as described above. The astable multivibrator outputs a train of pulses having a period determined based on the resistance of the resistor element 55 and the capacitance of the capacitor 58. Each pulse is produced as a reset pulse, which activates the resetting analog switch 45 of the peak holding circuit 40.

In the peak holding circuit 40, the peak holding capacitor 44 holds the peak values of the input signals but is reset when the analog switch 45 is made conductive in response to the reset pulse supplied from the reset pulse generator circuit 50. Thus, the period of the reset pulse is equal to a peak detection cycle. That is, a pulse output period of the astable

multivibrator is set so as to be equal to a desired peak detection period.

It is desired that this period is selected from 5ms to 100ms. This reason is that each audio signal corresponding to an object for analyzing the peak value thereof represents a low audio-range frequency component. The component is obtained by the low-pass filter 4 having a cut-off frequency selected from 100Hz to 200Hz. Thus, an upper-limit frequency of each of the audio signals input to the peak analyzer 6 is at most 200Hz. The period of each input signal having the upper-limit frequency is 5ms. Upon analysis of the peak value of a waveform, one cycle of the waveform may be considered as an object. Therefore, the peak detection cycle may be set to 5ms or longer.

A lower-limit frequency of each of the audio signals is sufficient if taken as being 10Hz or so where signals produced from a digital medium like a compact disk are taken into consideration. The period of each audio signal having the lower-limit frequency is 100ms. Thus, the upper limit of the peak detection cycle may be 100ms. The upper limit is actually set to a proper value selected from 5ms to 100ms taking into consideration the kind and cost of a music recording medium to be handled.

FIG. 12 is a block diagram showing one example of a specific circuit configuration of the voltage-controlled amplifier 7. Even in this case, only one channel of stereo two channels is shown. As illustrated in FIG. 12, the detected output of the peak analyzer 6 is applied to a photoelectric transducer 61 such as a photocoupler through a resistor element 63 as an input S_C. The photoelectric transducer 61 comprises a light-emitting diode 611 and a light-receiving element 612. Further, the output of the low-pass filter 4 is supplied to a non-inversion input terminal of an operational amplifier 62 as an input S_L. The non-inversion input terminal of the operational amplifier 62 is grounded through a resistor element 65. An inversion input terminal of the operational amplifier 62 is grounded through a resistive element 64. The light-receiving element 612 of the photoelectric transducer 61 is electrically connected between both an output terminal and the inversion input terminal of the operational amplifier 62.

In the so-constructed voltage-controlled amplifier 7, the intensity of light emitted from the light-emitting diode 611 varies depending on the output detected by the peak analyzer 6. The resistance of the light-receiving element 612 varies correspondingly.

That is, the value of a feedback resistance of the operational amplifier 62 changes. Thus, an amplification factor varies depending on the detected output of the peak analyzer 6. Described specifically, the amplification factor decreases as the detected output indicative of the peak value increases and hence the waveform is by-level compressed. Further, the output of the operational amplifier 62 is supplied to the mixture and distribution circuit 8 as the output of the voltage-controlled amplifier 7. The mixture and distribution circuit 8 combines the output signals of the voltage-controlled amplifier 7 which are used for the right R and left L channels and the output signals for the right R and left L channels, of the high-pass filter 5 and distributes the combined L signal to the power amplifiers 31 and 33 and the combined R signal to the power amplifiers 32 and 34. The respective power amplifiers 31 through 34 effect power amplification on the input signals and output the amplified signals to their corresponding speakers 21 through 24.

The operation of the audio signal processing circuit will next be described with reference to specific waveforms shown in FIG. 13. FIG. 13(A) shows, as an example, the manner of a change in waveform at the time that an audio signal for a rock has been

electrically processed. A feature of rock music is that a bass instrument is often used and the rock audio signal includes a number of low audio-range frequency components of 200Hz or less. That is, the waveform of the audio signal is represented as a waveform a, for example. When the signal having this waveform a is input to the low-pass filter 4, the low-pass filter 4 produces a signal having a waveform b. Further, the high-pass filter 5 generates a signal having a waveform c.

The peak analyzer 6 analyzes the waveform b and produces its peak value. The voltage-controlled amplifier 7 compresses the waveform b according to the peak value. In the example shown in FIG. 13(A), the signal having the waveform b is by-level compressed to about one-half the level of the signal, which is brought into a signal represented in the form of a waveform d. Incidentally, the waveform c corresponding to a high-frequency component is represented as an unprocessed waveform e. The mixture and distribution circuit 8 mixes the signal having the waveform d subjected to the waveform compression and that having the waveform e corresponding to the high-frequency component to produce a composite signal having a desired waveform f.

FIG. 13(B) shows, as an example, the manner of a change in waveform of an audio signal used in the pops music. A feature of the pops music is that a vocal range is occupied for the most part and a heavy bass portion is not extremely great or deep as indicated by a waveform g. Thus, the peak value of a signal having a waveform h, which has passed through the low-pass filter 4, is not so large. The voltage-controlled amplifier 7 outputs a signal having a waveform j, which is substantially identical to the input signal thereto. Further, the signal represented in the form of the waveform c is output to the high-pass filter 5 and a signal having an unprocessed waveform k is input to the mixture and distribution circuit 8. The mixture and distribution circuit 8 mixes the signal having the waveform j and the signal having the waveform k corresponding to a high-frequency component to produce a composite signal represented in the form of a waveform m. However, the waveshape of the produced signal is identical to the waveform g of the signal input to each of the low-pass filter 4 and the high-pass filter 5.

FIG. 13(C) illustrates, as an example, the manner of a change in waveform of an audio signal for the pops music subjected to low audio-range boosting. In this case, the low audio range is emphasized by the

audio signal processing circuit 2. When the user turns up the volume while the vehicle is running, for example, an audio signal whose level has been increased by the volume controller 3, is produced. That is, a signal having a waveform n higher in level than the waveform q is supplied to the low-pass filter 4 and the high-pass filter 5.

Since the low audio range is emphasized by the audio signal processing circuit 2, the low-pass filter 4 produces a signal represented in the form of a waveform o. Thus, the peak analyzer 6 analyzes the waveform o and produces its peak value. Further, the voltage-controlled amplifier 7 compresses the waveform o according to the peak value. In the example shown in FIG. 13(C), the signal having the waveform o is compressed in level to some certain degree, which is brought into a signal represented in the form of a waveform q. Further, the high-pass filter 5 produces a signal having a waveform p and the mixture and distribution circuit 8 is supplied with an unprocessed waveform r. The mixture and distribution circuit 8 mixes the signal having the waveform q subjected to the waveform compression and the signal of the waveform r corresponding to the high-frequency component so as to produce a composite signal having a desired waveform s.

As described above, the audio signal processing apparatus effects processes adapted to the input respective audio signals regardless of the presence of a preprocess such as the low audio-range boosting even if music sources vary in different form. Thus, unnecessary waveform compression is no longer effected on the audio signals used for the pops and classic music. The conventional audio signal processing apparatus could not carry out processes suitable for respective audio signals since signals falling within the whole audio range were subjected to waveform compression as objects to be by-level detected.

[Second Embodiment]

FIG. 14 is a schematic view showing the structure of an audio signal processing apparatus according to a second embodiment of the present invention. In the audio signal processing apparatus shown in FIG. 10, each of signals to be amplified by a voltage-controlled amplifier 7 is delayed one analysis cycle as compared with each of signals whose peak values are of objects to be analyzed by a peak analyzer 6. When each signal to be amplified by the voltage-controlled amplifier 7 coincides with the signal whose peak value is of the object to be analyzed by the peak

analyzer 6, a more excellent peak-value analysis effect can be brought about.

The audio signal processing apparatus shown in FIG. 14 has a structure which allows a peak-value analysis to be more effective. That is, delay portions 15a and 15b are added to the structure shown in FIG. 10 so as to fall between a low-pass filter 4 and the voltage-controlled amplifier 7. Further, delay portions 15c and 15d are provided between a high-pass filter 5 and a mixing circuit 11. The amount of delay of the respective delay portions 15a through 15d shows the amount corresponding to one analysis period of the peak analyzer 6. Incidentally, a description will be made herein of a two-speaker system as an example. Thus, the mixing circuit 11 is provided as an alternative to the mixture and distribution circuit 8 shown in FIG. 8. Further, the delay portions 15a through 15d are practical examples of delaying means.

The operation of the audio signal processing apparatus will now be described. An audio playback device 1 reproduces stereo signals from an audio recording medium and outputs the signals to an audio signal processing circuit 2. The audio signal processing circuit 2 effects signal processing on the input stereo signals in accordance with a low audio-

range boosting function, a graphic equalizer function for correcting an acoustic characteristic in a vehicle room and an auto-loudness function for varying a frequency characteristic in cooperation with the volume control effected with a view toward correcting a loudness characteristic on the mentality of hearing. Further, the audio signal processing circuit 2 outputs the so-processed signals to a volume controller 3. The volume controller 3 effects volume control processing on the input signals in accordance with a user's volume setting operation.

The output of the volume controller 3 is sent to the low-pass filter 4 and the high-pass filter 5. The low-pass filter 4 allows the passage of low audio-range frequency components of the signals input from the volume controller 3. The cut-off frequency of the low-pass filter 4 is set to one of 100Hz through 200Hz in a manner similar to the first embodiment. The high-pass filter 5 has a characteristic held in a supplementary relationship to a characteristic of the low-pass filter 4 and allows the passage of high audio-range frequency components of the signals input from the volume controller 3.

The L signal of the signals output from the low-pass filter 4 is sent to the peak analyzer 6 and the

delay portion 15a. The R signal of the output signals is supplied to the peak analyzer 6 and the delay portion 15b. The peak analyzer 6 detects the peaks of the signals output from the low-pass filter 4 at a time interval selected from 5ms through 100ms in a manner similar to the first embodiment. Further, the peak analyzer 6 compares the peak level of each of the input signals with a predetermined level and outputs a control voltage corresponding to the degree or rate of waveform compression depending on the result of comparison to the voltage-controlled amplifier 7. Incidentally, a detailed configuration and an operation of the peak analyzer 6 are identical to those of the peak analyzer 6 employed in the first embodiment and their detailed description will therefore be omitted herein.

The delay portions 15a and 15b respectively delay the signals output from the low-pass filter 4 one analysis period and output the same to the voltage-controlled amplifier 7. In response to the control voltage, the voltage-controlled amplifier 7 by-level compresses the signals output from the delay portions 15a and 15b in accordance with the control voltage. Further, the voltage-controlled amplifier 7 outputs the compressed signals to the mixing circuit 11. Incidentally, a detailed configuration and an operation

of the voltage-controlled amplifier ' ~e also identical to those of the voltage-controlled a ifier 7 employed in the first embodiment, and their detailed description will therefore be omitted herein.

The delay portions 15c and 15d respectively delay the signals output from the high-pass filter 5 one analysis cycle and output the same to the mixing circuit 11. The mixing circuit 11 combines the output signals of the voltage-controlled amplifier 7 which are used for the right R and left L channels and the output signals of the delay portions 15c and 15d. Further, the mixing circuit 11 outputs the combined L and R signals to their corresponding power amplifiers 31 and 32. The respective power amplifiers 31 and 32 effect power amplification on the input signals and output the amplified signals to their corresponding speakers 21 and 22.

Thus, the matching between the result of the waveform analysis and the audio signals to which the result of the waveform analysis is applied can be made by delaying the audio signals by the time interval required to effect the waveform analysis, thus making it possible to provide a reproduced sound of less abnormal or improper pitch.

In the audio signal processing apparatuses shown in FIGS. 10 and 14, the peak analyzer 6 and the voltage-controlled amplifier 7 are provided at a stage subsequent to the audio signal processing circuit 2 and the volume controller 3. Therefore, the SN ratio is slightly reduced. It is also considered that the peak analyzer 6 and the voltage-controlled amplifier 7 are provided at a stage prior to the audio signal processing circuit 2 and the volume controller 3 to prevent a reduction in the SN ratio. However, the detection of each peak value and the waveform compression, which are executed in view of audible volume conditions, cannot be carried out in this condition. Accordingly, the object of this invention cannot be achieved.

[Third Embodiment]

FIG. 15 is a schematic view showing the structure of a vehicle audio system including an audio signal processing apparatus according to a third embodiment of the present invention. In the same drawing, there are shown an electric volume 3A for amplifying audio signals under the control of a microcomputer and outputting the same to a distribution circuit 13, the microcomputer 9 having an A-D converter incorporated therein, for generating a volume control signal C_v and a voltage control signal V_r corresponding

to the volume control signal C_v, a multiplier 10 for multiplying a signal V_p produced from a peak analyzer 6 by the voltage control signal V_r produced from the microcomputer 9, a mixing circuit 11, a low audio-range boosting circuit 12 for effecting a bass boosting process on signals output from the mixing circuit 11 in response to the volume control signal C_v output from the microcomputer 9 and outputting the so-processed signals to the electronic volume 3A, and a control unit 14 for inputting a volume control instruction and the like by a user.

Incidentally, the microcomputer 9 and the electronic volume 3A show a practical example of a volume adjusting means. The multiplier 10 illustrates a practical example of a compression degree controlling means.

The operation of the audio signal processing apparatus will now be described. An audio playback device 1 reproduces stereo signals from an audio recording medium and outputs the same to a low-pass filter 4 and a high-pass filter 5. The low-pass filter 4 allows the passage of low audio-range frequency components of input signals. The high-pass filter 5 allows the passage of high audio-range frequency components of the input signals.

The peak analyzer 6 is activated in the same manner as that employed in the first embodiment so as to output the signal V_p having the voltage corresponding to the peak value. This signal V_p is then input to the multiplier 10. The multiplier 10 multiplies the signal V_p supplied from the peak analyzer 6 by the control signal V_r produced from the microcomputer 9 and outputs the result of multiplication to a voltage-controlled amplifier 7 as a control voltage corresponding to the rate of waveform compression. The voltage-controlled amplifier 7 is activated in the same manner as that employed in the first embodiment so as to compress the waveform of each of the audio signals falling within the low audio range.

As is understood from the specific configuration shown in FIG. 12, the voltage-controlled amplifier 7 is activated in such a manner that the amplification factor decreases with an increase in the control voltage. The control signal V_r output from the microcomputer 9 is set so as to correspond to a volume instruction value input by the user. Thus, even if there is no variation in the signal V_p output from the peak analyzer 6, the degree of the waveform compression increases if the volume instruction value input by the user is large. If the signal V_p output from the peak

analyzer 6 is large in level, that is, if the peak value of each of the audio signals falling within the low audio range is large even when there is no change in the volume instruction value input by the user, then the degree of the waveform compression is large. If the peak value of each audio signal falling within the low audio range is large and the volume instruction value input by the user is large, then the degree of the waveform compression becomes larger.

The mixing circuit 11 mixes the output of the voltage-controlled amplifier 7 and the output of the high-pass filter 5 and then supplies the composite output to the low audio-range boosting circuit 12. The low audio-range boosting circuit 12 is constructed as shown in FIG. 16, for example. FIG. 16 illustrates a low audio-range boosting circuit including an auto loudness correction function. Incidentally, only one channel is illustrated in this example. However, the low audio-range boosting circuit is actually provided for each of the left and right two channels.

In the low audio-range boosting circuit 12 shown in FIG. 16, an audio signal L_{in} output from the mixing circuit 11 is supplied to a non-inversion input terminal of an operational amplifier 71 which serves as a so-called buffer amplifier. The output of the

operational amplifier 71 is sent to a non-inversion input terminal of an operational amplifier 72 through a resistor element 75 and fed back to an inversion input terminal of the operational amplifier 71.

The output of the operational amplifier 72 is supplied to the electronic volume 3A as the output of the low audio-range boosting circuit 12 and fed back to an inversion input terminal of the operational amplifier 72 through a resistor element 77. An analog switch 73 supplied with a volume control signal C_v output from the microcomputer 9 as a control input is provided between the inversion input terminal and the non-inversion input terminal of the operational amplifier 72. The analog switch 73 comprises a movable terminal 73a grounded through a series body comprised of a resistor element 74 and an inductor 76, and respective fixed terminals 73b drawn from a plurality of taps of resistor elements inserted between the inversion input terminal and the non-inversion input terminal of the operational amplifier 72. A low audio-range boost characteristic determined in accordance with the resistance and inductance corresponding to the state of switching action of the analog switch 73 and a negative feedback resistance based on a resistive element 77, is applied

to the audio signals supplied to the operational amplifier 72.

The microcomputer 9 outputs the volume control signal C_v indicative of the volume instruction value input by the user to the low audio-range boosting circuit 12. The analog switch 73 of the low audio-range boosting circuit 12 varies the position of the movable terminal 73a depending on the value of the volume control signal C_v. Thus, the low audio-range boost characteristic varies due to a change in the resonance resistance. The output of the low audio-range boosting circuit 12 is supplied to the electronic volume 3A.

The electronic volume 3A amplifies the input signals in accordance with the value of the volume control signal C_v output from the microcomputer 9 and outputs the amplified signals to the distribution circuit 13. The distribution circuit 13 distributes L and R signals supplied from the electronic volume 3A to their corresponding power amplifiers 31 and 33, and 32 and 34. The respective power amplifiers 31 through 34 increase the power of the input signals and output the same to respective speakers 21 through 24.

Thus, when the product of the signal V_p produced from the peak analyzer 6 and the control signal V_r is large, the degree of reduction in the

amplification factor of the voltage-controlled amplifier 7 increases. When, on the other hand, the product is small, the degree of reduction in the amplification factor decreases. That is, when the peak value of a low audio-range frequency component of each audio signal is large and the volume control signal is high in level, the degree of the waveform compression is brought into a higher state. Even in the case of the audio signal processing apparatus according to the present embodiment, a process adapted to a music genre can be realized that the waveform compression is made to a music source such as a rock music source, whose low audio-range components are excessively high in level and the waveform compression is not made to a classic music source. In this case as well, a process including user's volume control instructions too can be materialized. Further, since the electronic volume 3A for effecting the volume control is provided at a subsequent stage of the entire circuit, a reduction in the SN ratio of the audio signal, which occurs due to the existence of the audio signal processing apparatus, is small.

[Fourth Embodiment]

FIG. 17 is a schematic view showing the structure of an audio signal processing apparatus

according to a fourth embodiment of the present invention. In the present embodiment, delay portions 15a and 15b are added to the structure shown in FIG. 15 so as to fall between the low-pass filter 4 and the voltage-controlled amplifier 7. Further, delay portions 15c and 15d are additionally provided between the high-pass filter 5 an the mixing circuit 11. The respective delay portions 15a through 15d are identical to those shown in the second embodiment. The amount of delay of the delay portions 15a through 15d represents an amount corresponding to one analysis period of the peak analyzer 6. Incidentally, a description will be made herein of a two-speaker system as an example. Thus, the distribution circuit 13 shown in FIG. 15 is not provided in the present embodiment.

The operation of the audio signal processing apparatus will next be described. An audio playback device 1 reproduces stereo signals from an audio recording medium and outputs the same to a low-pass filter 4 and a high-pass filter 5. The low-pass filter 4 allows the passage of low audio-range frequency components of the input signals. The high-pass filter 5 allows the passage of high audio-range frequency components of the input signals.

A peak analyzer 6, a microcomputer 9 and a multiplier 10 are activated in the same manner as those employed in the third embodiment. The multiplier 10 outputs the result of multiplication to a voltage-controlled amplifier 7 as a control voltage corresponding to the degree of waveform compression. The voltage-controlled amplifier 7 is operated in the same manner as that employed in each of the aforementioned embodiments so as to compress the waveform of each audio signal which falls within a low audio range.

Similarly to the second embodiment, the delay portions 15a and 15b delay the output signals of the low-pass filter 4 one analysis period so as to be output to the voltage-controlled amplifier 7. In response to the control voltage supplied from the multiplier 10, the voltage-controlled amplifier 7 compresses the level of each of signals output from the delay portions 15a and 15b in accordance with the control voltage. The delay portions 15c and 15d delay signals output from the high-pass filter 5 one analysis period so as to be output to the mixing circuit 11.

The subsequent operation is identical to that effected in the third embodiment and its description will therefore be omitted herein. The audio signal

processing apparatus according to the present embodiment can bring about the same effect as that obtained by the audio signal processing apparatus according to the third embodiment. Further, the matching between the result of waveform analysis and the audio signals to which the result of the waveform analysis is applied can be made by delaying the audio signals by the time interval required to conduct the waveform analysis, thus making it possible to provide a reproduced sound of less abnormal or improper pitch.

[Fifth Embodiment]

FIG. 18 is a schematic view showing the structure of an audio signal processing apparatus according to a fifth embodiment of the present invention. In the audio signal processing apparatuses according to the aforementioned embodiments, waveform distortion is extremely low and the quality of tones remains unchanged in an audio range other than a low audio range. When, however, the low-pass filter 4 and the high-pass filter 5 made by analog circuits are used, the accuracy of division of a frequency band is lowered due to variations in constants of circuit parts, with the result that the tone quality is often deteriorated. If, on the other hand, the division of a frequency band and the compression of each waveform are carried out in

accordance with a digital signal process using a digital circuit such as a digital filter, then the accuracy of processing depends on an operational word length but a process which is free of variations and high in accuracy, can be achieved. The audio signal processing apparatus according to the present embodiment has been made on the basis of such an idea as referred to above.

Referring now to FIG. 18, reference numeral 80 indicates an A-D converter for effecting A-D conversion on audio signals output from an audio playback device 1. Designated at numeral 81 is a digital band dividing circuit having two quadrative mirror filters (QMFs) 81a and 81b, for dividing a frequency band. Here, the QMF 81a is used to divide a band of frequencies of an L signal, whereas the QMF 81b is used to divide a frequency band of an R signal. Reference numeral 88 indicates a digital peak detector for detecting the peak value of each low audio-range frequency component in accordance with a digital process. Designated at numeral 89 is a digital waveform compressor for effecting waveform compression on the input signals in accordance with the digital process.

There are also shown a digital mixing circuit 83 having two QMFs 83a and 83b, for mixing the output of the digital waveform compressor 89 and the outputs of

the QMFs 81a and 81b, a digital processing circuit 85 for effecting processes such as a volume control process, a distribution process, etc. on the output of the digital mixing circuit 83, and D-A converters 86 and 87 for respectively effecting D-A conversion on the outputs of the digital processing circuit 85 and supplying the results of conversion to their corresponding power amplifiers (not shown). Designated at numeral 90 is a controller for effecting a process such as the supply of a volume control signal to the digital processing circuit 85. In the present embodiment, a four-speaker system is described as an example. The present audio signal processing apparatus corresponds to one of a type wherein the audio signal processing apparatus according to the first embodiment has been designed in digital form. It is however a point of importance that the QMFs are used for the band division process and the mixing process.

Incidentally, the digital band dividing circuit 81 is of a practical example of a band-dividing filter means and the digital peak detector 88 shows a practical example of a waveform analyzing means. Further, the digital waveform compressor 89 illustrates a typical configuration of a waveform compressing means

and the digital mixing circuit 83 is of a practical example of a mixing means.

The operation of the audio signal processing apparatus will now be described. The audio signals output from the audio playback device 1 are first A-D converted by the A-D converter 80 and then supplied to the digital band dividing circuit 81. In the digital band dividing circuit 81, the QMF 81a divides a frequency band of an L signal of the audio signals. Further, the QMF 81a supplies a low frequency output to the digital peak detector 88 and sends a high frequency output to the digital mixing circuit 83. On the other hand, the QMF 81b divides a frequency band of an R signal of the audio signals.

FIG. 19 shows a specific configuration of each of the QMFs 81a and 81b. As shown in FIG. 19, each of the QMFs 81a and 81b is of a symmetric coefficient FIR filter for dividing a band into two and comprises n (n: odd number) cascade-connected delay portions 101_1 through 101_n, multipliers 103_0 and 103_n electrically connected to both an input terminal of each QMF and an input terminal of the delay portion 101_1, and an output terminal of the delay portion 101_n respectively and multipliers 103_1 through 103_{n-1} electrically connected to their corresponding output terminals of the delay

portions 101_1 through 101_{n-1} and their corresponding input terminals of the delay portions 101_2 through 101_n, adders 102_2, 102_4, ..., 102_n for adding odd-numbered multiplied outputs (odd-numbered tap outputs) together, adders 102_1, 102_3, ..., $102_{(n-1)}$ for adding even-numbered multiplied outputs (even-numbered tap outputs) together, an adder 102_H for subtracting the output of the adder 102_{n-1} from the output of the adder 102_n to produce a high frequency output, and an adder 102_L for adding together the output of the adder $102_{(n-1)}$ and the output of the adder 102_n to produce a low frequency output.

Here, multiplying factors of the multipliers 103_0 through 103_n are represented as a_0, a_1, ..., a_n in that order. The relationship of $a_{n-1} = a_i$ (i: natural number) is met between the multiplying factors.

In the QMFs constructed as described above, the high frequency output and the low frequency output can be simultaneously obtained by separately effecting the addition of the odd-numbered tap outputs and the addition of the even-numbered tap outputs and taking every other one out of the respective added outputs. The low frequency output is sent to the digital peak detector 88 and the high frequency output is supplied to the digital mixing circuit 83.

The digital peak detector 88 produces a control signal in accordance with the same process as that effected by the peak analyzer 6 shown in FIG. 10. However, this process is carried out as the digital process. The digital waveform compressor 89 outputs the low audio-range frequency components which have been compressed in accordance with the process similar to that by the voltage-controlled amplifier 7 shown in FIG. 10. In this case, this process is based on the digital process.

The QMF 83a in the digital mixing circuit 83 mixes the low audio-range frequency components of the L signal output from the digital waveform compressor 89 and the high audio-range frequency components of the L signal output from the QMF 81a. Further, the QMF 83b mixes the low audio-range frequency components of the R signal output from the digital waveform compressor 89 and the high audio-range frequency components of the R signal output from the QMF 81b.

FIG. 20 shows a specific configuration of each of the QMFs 83a and 83b. As shown in FIG. 20, each of the QMFs 83a and 83b comprises an adder 101_H for adding the low frequency output produced from the digital waveform compressor 89 to the inversed of the high frequency output produced from the digital band dividing

circuit 81, delay portions 101_1 through 101_{n-1} electrically cascade-connected to the output of the adder 101_H, an adder 101_L for adding the high frequency output produced from the digital band dividing circuit 81 to the low frequency output produced from the digital waveform compressor 89, delay portions 101_1 through 101_n electrically cascade-connected to the output of the adder 101_L, multipliers 103_0 through 103_n respectively electrically connected to the output of the adder 101_H and the alternate outputs of the delay portions 101_1 through 101_n, adders 102_1 through 102_n for adding respective multiplied outputs together, and a coefficient multiplier 104 for multiplying the value added by the adders 102_1 through 102_n by 1/2.

Multiplying factors of the multipliers 103_0 through 103_n are represented as a_0, a_1, ..., a_n in that order. The relationship of $a_{n-1} = a_i$ (i: natural number) is met between the multiplying factors.

If "0" is interpolated into the alternate input digital signal and a signal represented in a sampling frequency which has been brought to an initial condition is input to each of the thus-constructed QMFs 83a and 83b, then a digital signal whose high audio-range frequency components and low audio-range frequency components have been combined together is reproduced.

The digital processing circuit 85 effects the volume control and the like on the signals output from the digital mixing circuit 83 and then distributes the so-processed signals to the D-A converters 86 and 87. Each of the D-A converters 86 and 87 D-A converts the output of the digital processing circuit 85 and supplies it to each of power amplifiers (not shown).

Since the QMFs symmetrical with respect to a filter for dividing the frequency band of each audio signal and a filter for combining a high audio-range frequency component and a low audio-range frequency component together are used as described above, a signal process capable of improving the accuracy of signal reproduction can be realized. With this arrangement, the compression process is not effected on signals such as pops and classic audio signals, which do no cause excessive variations in frequency spectrum. Further, deterioration in the signals is not produced upon execution of the band division and the synthesis of the low and high audio-range frequency components. Therefore, the audio signals can be reproduced with fidelity like the signal processing circuit does not exist. When an analog filter is used as the filter for dividing the frequency band, there is often a situation in which the signals cannot be prevented from being

deteriorated upon execution of the band division and the combination of the low and high audio-range frequency components even when the compression process is not executed. Therefore, the reproduced sound is often deteriorated in some degree.

[Sixth Embodiment]

In the audio signal processing apparatus according to the fifth embodiment, it is necessary to provide a number of delay portions and multipliers as shown in FIGS. 19 and 20. Therefore, much time is required for filter processes. It is expected that a processed input signal is delayed 100ms or longer, for example.

Thus, when it is desired to preferentially obtain a simplified circuit even if the fidelity of waveform transmission is sacrificed more or less, a IIR filter show in FIG. 21 may be used as an alternative to each of the QMFs 81a and 81b. Incidentally, the entire structure of the fifth embodiment corresponds to the structure of the audio signal processing apparatus shown in FIG. 18 from which the QMFs 81a, 81b, 83a and 83b have been removed.

As shown in FIG. 21, delay portions 101_0 through 101_4, multipliers 103_1 through 103_4 and an adder 102 are substantially reduced in number. Thus, an

operating time period is also greatly reduced. For example, the operating time can be set to a short time interval of less than or equal to 100ms. The digital mixing circuit 85 can be replaced by a simple adder. Thus, the scale and cost of a digital signal processing circuit can be substantially reduced.

[Seventh Embodiment]

FIG. 22 is a schematic view showing the structure of an audio signal processing apparatus according to a seventh embodiment of the present invention. The audio signal processing apparatus corresponds to one of a type wherein digital delay portions 82a through 82d are additionally disposed in the audio signal processing apparatus shown in FIG. 18 so as to fall between the digital band dividing circuit 81 and both the digital waveform compressor 89 and the digital mixing circuit 83 of the audio signal processing apparatus shown in FIG. 18. The amount of delay of the digital delay portions 82a through 82d is set so as to be equal to a processing time interval (i.e., a time interval from the input of signals to the output of the result of processing) of a digital peak detector 88.

Operations such as the division of a frequency band by a digital band dividing circuit 81, a peak detecting process of the digital peak detector 88, a

waveform compression process of a digital waveform compressor 89 and a mixing process of a digital mixer 83 are identical to those executed in the audio signal processing apparatus according to the fifth embodiment. Further, the digital delay portions 82a through 82 e activated in the same manner as the de..y portions .5a through 15d employed in each of the audio signal processing apparatuses according to the second and fourth embodiments. Thus, the matching between the result of waveform analysis and the audio signals to which the result of waveform analysis is applied can be made in a manner similar to the second and fourth embodiments, thus making it possible to provide a reproduced sound of less improper pitch.

[Eighth Embodiment]

FIG. 23 is a schematic view showing the structure of an audio signal processing apparatus according to an eighth embodiment of the present invention. In the audio signal processing apparatuses shown in FIGS. 18 and 22, the digital band dividing circuit 81, the digital peak detector 88, the digital waveform compressor 89 and the digital mixing circuit 85 have been respectively constructed in the form of discrete digital circuits. In the audio signal processing apparatus illustrated in FIG. 22 as well, the

digital delay portions 82a through 82d have been respectively constructed in the form of discrete digital circuits. On the other hand, in the audio signal processing apparatus according to the present embodiment, these circuits are integrally constructed in the form of a digital signal processor (hereinafter called "DSP").

A DSP 100 executes functions of the digital band dividing circuit 81, the digital peak detector 88, the digital waveform compressor 89, the digital mixing circuit 85 and the digital delay portions 82a through 82d in accordance with a procedure based on a program. Accordingly, the operation of the audio signal processing apparatus is identical to that of the audio signal processing apparatus shown in FIG. 18 according to the fifth embodiment and that of the audio signal processing apparatus shown in FIG. 22 according to the seventh embodiment. Therefore, the detailed description of the operation will be omitted in the present embodiment.

Thus, the respective digital circuits are simplified by integrating digital processes into the DSP 100. Further, if a controller 90 comprised of a microcomputer effects control for adjusting processes other than the waveform compression process, such as

volume control, equalizing, etc., and the DSP 100 also executes adjusting processes under the control of the controller 90, then the circuits can further be simplified. When the digital delay portions 82a through 82d are not used, that is, when the audio signal processing apparatus according to the fifth embodiment shown in FIG. 18 is brought into integration using the DSP 100, it is of course unnecessary to set a program for realizing the functions of the digital delay portions 82a through 82d to the DSP 100.

[Ninth Embodiment]

FIG. 24 is a schematic view showing the structure of an audio signal processing apparatus according to a ninth embodiment of the present invention. In the audio signal processing apparatus, the digital delay portions 82a through 82d can be achieved by a signal delay processing means 101 corresponding to a memory provided outwardly of a DSP 100. Thus, the DSP 100 executes respective functions of a digital band dividing circuit 81, a digital peak detector 88, a digital waveform compressor 89 and a digital mixing circuit 85, i.e., arithmetic processing functions (processes other than a delay function) alone. Thus, a program for the DSP 100 can be reduced by separating the entire functional portion into a portion

for realizing the arithmetic processing functions and a portion for realizing the delay function. As a result, the DSP 100 can be prevented from being constructed on a large scale. Thus, the entire circuit scale is reduced and hence a more suitable audio signal processing apparatus can be obtained. Incidentally, the capacity of the memory constituting the signal delay processing means (i.e., memory) 101 may be set so as to correspond to the capacity for storing a waveform sample at one analysis period corresponding to a time length required to analyze the peak value, practically, at a time interval falling within a time range from 5ms to 100ms.

The contents of the digital signal processing from the beginning of analysis of the peak value to the completion of waveform compression will next be described specifically. Now, consider where the following processes are applied to the audio signal processing apparatus using the DSP 100 and the memory 101 shown in FIG. 24. It is however needless to say that the following processes can be applied to the audio signal processing apparatus using the DSP 100 shown in FIG. 23. Further, the following processes can also be applied even to the audio signal processing apparatus shown in FIG. 22. If a temporary storing process (delaying process) in the following processes is

skipped, then the following processes can also be applied even to the audio signal processing apparatus shown in FIG. 18.

The overall flow of the digital signal processing will first be described in accordance with a flowchart shown in FIG. 25. When an audio playback device 1 starts reproducing, an A-D converter 80 effects A-D conversion on audio signals and outputs the so-processed signals as sample waveform data. The DSP 100 takes in the waveform data (Step ST11) and effects a band dividing process using QMF and the like (Step ST12). Then, waveform data about low and high audio-range frequency components which have been subjected to the band division, are written into the memory 101 (Steps ST13 and ST14). In the audio signal processing apparatus shown in FIG. 22, the digital band dividing circuit 81 effects the processes of Steps ST11 and ST12. The waveform data supplied from the A-D converter 80 are sent to the digital delayers 82a and 82d.

The DSP 100 effects peak-value analysis on the respective waveform data about the low audio-range frequency components (Step ST15) and calculates a compression degree corresponding to each of the peak values indicative of the result of analysis (Step ST16). In the audio signal processing apparatus shown in FIG.

22, the digital peak detector 88 effects the processes of Steps ST15 and ST16.

Next, the DSP 100 effects a compression process corresponding to the compression rate on the waveform data about the low audio-range frequency components stored in the memory 101 (Step ST17). In the audio signal processing apparatus shown in FIG. 22, the digital waveform compressor 83 effects the process of Step ST17 on the waveform data supplied from the digital delay portions 82a and 82b.

Further, the DSP 100 mixes the compressed waveform data and the waveform data about the high audio-range frequency components stored in the memory 101 using the QMF and the like (Step ST18). Thereafter, the DSP 100 outputs the mixed waveform data to a D-A converter 86 (Step ST19). In the audio signal processing apparatus shown in FIG. 22, the digital mixing circuit 83 effects the processes of Steps ST18 and ST19 on the compressed waveform data and the waveform data about the high audio-range frequency components the latter of which have been supplied from the digital delay portions 82c and 82d.

Here, the respective waveform data, which have been subjected to the band division, are temporarily stored in the memory 101 or delayed by the digital delay

portions 82a through 82d. Accordingly, the waveform data are placed in a waiting state by a time interval from the beginning of the peak-value analysis to the beginning of the waveform compression. Thus, the waveform data to be compressed can be surely compressed. A method of placing such data to be processed in the waiting state is well known in, for example, a speech analysis and synthesis field as a sort of predictive analysis method. However, if a predictive analysis method is applied to a signal processing system which forms or contributes to the optimization of an audio reproduction system, then a peculiar effect that the conventional system cannot obtain can be produced.

The time transition of signals having specific waveforms produced in accordance with the aforementioned processes will now be described with reference to waveform charts shown in FIG. 26. Now, consider that A-D converter 80 outputs waveform data shown in FIG. 26(a) from a sample waveform take-in start point (t=0) as the starting point. Thus, waveform data about low audio-range frequency components shown in FIG. 26(b) and waveform data about high audio-range frequency components shown in FIG. 26(c) are both obtained in accordance with the band dividing process. Let's now

assume that a time interval τ_1 is taken to effect the band dividing process.

Then, the peak-value analysis is carried out. The waveform data about the low audio-range frequency components in one frame, as shown in FIGS. 26(i) and 26(j), are regarded as objects at the analysis. In this case, processing period is between τ_1 and τ_2. A compression degree calculation process is executed in each timing shown in FIG. 26(k) depending on the result of analysis.

As shown in FIG. 26(d), the memory 101 or the digital delay portions 82a and 82b produce waveform data about low audio-range frequency components, which have been delayed one frame. The waveform compression is effected on these waveform data in each timing shown in FIG. 26(l). The waveform compression is executed for each waveform data. The compressed waveform data are produced as illustrated in FIG. 26(f). Further, waveform data about high audio-range frequency components, which have been delayed one frame, are output from either the memory 101 or the digital delay portions 82c and 82d as shown in FIG. 26(e). Thus, the waveform data about the low audio-range frequency components, which have been subjected to the waveform compression and the waveform data about the high audio-

range frequency components, which have been subjected to the waveform compression, are combined together in each timing shown in FIG. 26(m) to produce composite data shown in FIG. 26(g).

Subsequently, the above-described processes are repeated for each frame. The time interval τ_1 required to effect the band division is preferably as short as possible. When the QMFs shown in FIGS. 19 and 20 are used, much time is required. When, on the other hand, the IIR filter is used, the time interval τ_1 can be reduced to a 1/10 or less of the time interval τ_1 as compared with the case where the QMFs are used. In this case, a delay time of each audio signal from the audio playback device 1 to each power amplifier is substantially equal to that taken by the memory 101 and becomes at most 100ms or so.

[Tenth Embodiment]

A predictive analysis method of more effectively conducting the waveform compression will next be described specifically by examples shown in FIGS. 27 and 28. FIG. 27 is a flowchart for describing an analytical procedure, which corresponds to the analysis and compression process described in FIG. 25. FIG. 28 is a waveform diagram showing the waveforms of audio signals, wherein FIG. 28(a) shows a non-processed

waveform and FIG. 28(b) illustrates a processed waveform. Now, consider where this method is applied to the audio signal processing apparatus shown in FIG. 24. However, such a method can also be applied to the audio signal processing apparatus shown in FIG. 22 or FIG. 23.

The DSP 100 sets waveform data about a low audio-range frequency component from a zero point (or point near the zero point) of the waveform to the next zero point as data corresponding to one frame. That is, an time interval between adjacent zero points on a digital audio waveform is defined as an analysis time length. Thus, when a zero point is detected from the waveform data about the low audio-range frequency component, the DSP 100 starts to analyze the peak value in one frame.

Each zero point is easily obtained as a point inverting the sign of the sample waveform data. In the waveform diagram shown in FIG. 28, the analysis time length corresponds to each of time intervals between T_0 and T_1 and between T_1 and T_2, for example. The DSP 100 makes a decision as to whether there is waveform data exceeding a threshold level V_s with waveform data within one analysis time as an object (Step ST22). The threshold level V_s is defined as a value determined in advance based on a linear region characteristic of an

audio system and the like. The points of generation of the waveform data which exceed the threshold level V_s are represented as T_s points.

When the T_s points exist, a point at which the peak value is brought to the maximum (maximum represented in the form of the absolute value) is extracted from these points. As a result, a peak value V_p at the extracted point can be obtained (Steps ST23 and ST24). Then, a gain factor G corresponding to the compression degree is calculated as $|V_s/V_p|$ (Step ST25). When, on the other hand, there are no T_s points, the waveform compression is unnecessary and hence the gain factor G is set to 1 (i.e., G=1) (Steps ST23 and ST26). The processes of Steps ST22 through ST26 are executed by the digital peak detector 88 in the audio signal processing apparatus shown in FIG. 22.

After a stay to be described later has ended, the DSP 100 reads the waveform data of one frame about the low audio-range frequency component from the memory 101. Then, each waveform data is multiplied by the gain factor G to produce compressed waveform data about the low audio-range frequency component. Incidentally, the data stored in the memory 101 may be multiplied by the gain factor G so as to read compressed waveform data about low audio-range frequency components after the

stay. In the audio signal processing apparatus shown in FIG. 22, the process of Step ST27 is executed by the digital waveform compressor 83.

In the waveform shown in FIG. 28(a), the waveform data exceeding the threshold level V_s exist between T_2 and T_3 and the maximum value V_p is produced at a point T_p. The waveform data, which exceed the threshold level V_s, do not exist between the zero points other than T_2 and T_3. Accordingly, the waveform compression is executed between T_2 and T_3 alone. As shown in FIG. 28(b), the waveform data is output in which the maximum peak value between T_2 and T_3 has been brought to V_s.

By setting the time interval between the adjacent zero points to the analysis time length in this way, the continuity of the compressed waveform is maintained and waveform distortion is reduced. The waveshape shown in FIG. 28(b) is actually delayed by the stay in the memory 101 or the amount of delay of the digital delay portions 82a through 82d as compared with the waveshape shown in FIG. 28(a). When the lowest frequency component included in the audio signal is now regarded as 20Hz, for example, a time interval between adjacent zero points on a sine wave of that frequency becomes 25ms. Therefore, if the amount of delay is set

to 25ms, then there is no longer produced a frame which reaches or corresponds to a time interval exceeding the amount of the delay, thus causing no problem in the waveform analysis.

[Eleventh Embodiment]

In order to make a further reduction in the waveform distortion during the waveform compression process, the analysis time length may be a time interval between a certain zero point and the next zero point but one as shown in FIG. 29. That is, a time interval between adjacent zero points which are selected out of respective points at which the waveform crosses a zero level and which extend along the same cross direction (positive to negative or vice versa), is set as one frame. Incidentally, FIG. 29(a) shows a non-processed waveform and FIG. 29(b) illustrates a processed waveform. In the DSP 100 or the digital peak detector 88, waveform data between a zero point and the following zero point is regarded as an object for the detection of the peak value and the calculation of the gain factor. A specific procedure is identical to that executed in the tenth embodiment.

In this case, one interval for the compression of a waveform includes positive and negative regions of the waveform, that is, it corresponds to one wavelength.

Therefore, the interval is wider than that used in the tenth embodiment. However, harmonic distortion produced due to the waveform compression can be less reduced by setting the analysis time interval in this way. In this case, the capacity of the memory 101 is increased to about two times the capacity employed in the tenth embodiment.

[Twelfth Embodiment]

The waveform compression processes, which have been executed in the aforementioned embodiments, were all those based on a so-called variable length frame that the analysis time length varies according to the difference between configurations of each waveform. However, the analysis time interval may also be based on a fixed length frame. It is preferable that an analysis time interval of a fixed length corresponding to the fixed length frame falls within a range of from 5ms to 100ms as has already been described above. The contents of the waveform compression process will be described below with reference to a flowchart shown in FIG. 30 and a waveform chart illustrated in FIG. 31. Here, the audio signal processing apparatus shown in FIG. 23 will be described by way of example. However, such a waveform compressing method can be applied even to the

audio signal processing apparatuses shown in FIGS. 22 and 24.

When the audio playback device 1 starts reproducing, the DSP 100 starts processes such as a band dividing process, a peak-value analyzing process, etc. Incidentally, the DSP 100 sets up a threshold level V_s and sets "1" to a gain factor G_{-1} of a previous frame upon initialization. During the peak-value analyzing process, the DSP 100 receives waveform data corresponding to one frame from a band processing portion (Step ST31). Further, the DSP 100 detects a zero point from each waveform data (Step ST32). Incidentally, a start point of one frame is called a start endpoint and an end point thereof is called a terminal point. Further, the zero point is represented as T_i (where $i \geq 0$). The DSP 100 multiplies waveform data between the start endpoint and T_0 by the gain factor G_{-1} (Step ST33).

Next, the DSP 100 detects whether waveform data exceeding the threshold level V_s exist in waveform data between T_0 and the terminal point (Step ST34). The points at which the waveform data exceeding the threshold level V_s exist are respectively represented as T_s. If the T_s points exist, then the maximum value (maximum value represented in the form of the absolute

value) V_p is extracted from waveform data at these points (Step ST35 and ST37). Further, $|V_s/V_p|$ is set as a gain factor G corresponding to the compression rate (Step ST38). If there are no T_s points, "1" which means that the waveform compression is not made, is set as the gain factor G (Step ST36).

Then, each waveform data, which appears between the T_0 point and the terminal point, is multiplied by the gain factor G (Step ST39). Further, a gain factor G for the present frame is set as a gain factor G_{-1} used in the next frame (Step ST40).

When a waveform shown in FIG. 31(a), for example, is taken, a waveform illustrated in FIG. 31(b) is obtained in accordance with the aforementioned processes. That is, the waveshape from the T_0 point and the terminal point is compressed up to about 80%. The compression degree applied to the waveform between the T_0 point and the terminal point is applied to the waveform between the terminal point and the initial zero point T_1 of the next frame. The waveform shown in FIG. 31(b) is actually delayed in time as compared with the waveform shown in FIG. 31(a).

Thus, the continuity of each waveform can be held and the waveform distortion can be minimized in a manner similar to the above respective embodiments by

detecting the zero points and processing each waveform between the adjacent zero points at the same compression rate. Thus, a deterioration in tone quality is reduced. Further, since the analysis time length is of the fixed length, the delay time of the waveform data processed by the DSP 100 is identical to the analysis time length. Accordingly, the control regarding the input and output of the sample waveform data can be made uncomplicated and a circuit configuration can be simplified.

Having now fully described the invention, it will be apparent to those skilled in the art that many changes and modifications can be made without departing from the spirit or scope of the invention as set forth herein.

WHAT IS CLAIMED IS:

1. An audio signal processing apparatus comprising:

band-dividing filter means for by-band dividing an audio signal into a low audio-range signal and a high audio-range signal;

waveform analyzing means for analyzing the peak value of the low audio-range signal obtained by said band-dividing filter means;

waveform compressing means for compressing the level of a waveform of the low audio-range signal according to the result of analysis by said waveform analyzing means; and

mixing means for mixing the low audio-range signal compressed by said waveform compressing means and the high audio-range signal.

2. An audio signal processing apparatus according to claim 1, wherein said band-dividing filter means, said waveform analyzing means and said waveform compressing means are respectively made up of digital processing circuits.

3. An audio signal processing apparatus according to claim 2, wherein said band-dividing filter means comprises a digital mirror filter.

4. An audio signal processing apparatus according to claim 2, wherein said band-dividing filter means, said waveform analyzing means and said waveform compressing means are constructed in the form of a digital signal processor.

5. An audio signal processing apparatus according to claim 4, wherein said band-dividing filter means comprises a digital mirror filter.

6. An audio signal processing apparatus according to claim 1, further comprising volume adjusting means for adjusting the level of the audio signal and compression degree controlling means for reflecting the degree of control by said volume adjusting means in the result of analysis by said waveform analyzing means.

7. An audio signal processing apparatus according to claim 6, wherein said band-dividing filter means, said waveform analyzing means and said waveform compressing menas are respectively made up of digital processing circuits.

8. An audio signal processing apparatus according to claim 7, wherein said band-dividing filter means comprises a digital mirror filter.

9. An audio signal processing apparatus according to claim 7, wherein said band-dividing filter

means, said waveform analyzing means and said waveform compressing means are constructed in the form of a digital signal processor.

10. An audio signal processing apparatus according to claim 9, wherein said band-dividing filter means comprises a digital mirror filter.

11. An audio signal processing apparatus comprising:

band-dividing filter means for by-band dividing an audio signal into a low audio-range signal and a high audio-range signal;

waveform analyzing means for analyzing the peak value of the low audio-range signal obtained by said band-dividing filter means;

delaying means for supplying a delay of a time interval including a time interval required for the processing of analysis by said waveform analyzing means to the low audio-range signal and the high audio-range signal;

waveform compressing means for compressing the level of a waveform of said delayed low audio-range signal according to the result of analysis by said waveform analyzing means; and

mixing means for mixing the low audio-range signal compressed by said waveform compressing means and

the high audio-range signal delayed by said delaying means.

12. An audio signal processing apparatus according to claim 11, wherein said band-dividing filter means, said waveform analyzing means and said waveform compressing means are respectively made up of digital processing circuits.

13. An audio signal processing apparatus according to claim 12, wherein said band-dividing filter means comprises a digital mirror filter.

14. An audio signal processing apparatus according to claim 12, wherein the amount of delay by said delaying means falls within a range of 5ms to 100ms.

15. An audio signal processing apparatus according to claim 12, wherein said waveform analyzing means effects waveform analysis with a time inte. al between adjacent zero cross-points of each waveform as a unit.

16. An audio signal processing apparatus according to claim 12, wherein said waveform analyzing means effects waveform analysi ith each waveform within a fixed time interval a. unit.

17. An audio signal processing apparatus according to claim 12, wherein said band-dividing filter

means, said waveform analyzing means and said waveform compressing means are constructed in the form of a digital signal processor.

18. An audio signal processing apparatus according to claim 17, wherein said band-dividing filter means comprises a digital mirror filter.

19. An audio signal processing apparatus according to claim 11, further comprising volume adjusting means for adjusting the level of the audio signal and compression degree controlling means for reflecting the degree of control by said volume adjusting means in the result of analysis by said waveform analyzing means.

20. An audio signal processing apparatus according to claim 19, wherein said band-dividing filter means, said waveform analyzing means and said waveform compressing means are respectively made up of digital processing circuits.

21. An audio signal processing apparatus according to claim 20, wherein said band-dividing filter means comprises a digital mirror filter.

22. An audio signal processing apparatus according to claim 20, wherein the amount of delay by said delaying means falls within a range of 5ms to 100ms.

23. An audio signal processing apparatus according to claim 20, wherein said waveform analyzing means effects waveform analysis with a time interval between adjacent zero cross-points of each waveform as a unit.

24. An audio signal processing apparatus according to claim 20, wherein said waveform analyzing means effects waveform analysis with each waveform within a fixed time interval as a unit.

25. An audio signal processing apparatus according to claim 20, wherein said band-dividing filter means, said waveform analyzing means and said waveform compressing means are constructed in the form of a digital signal processor.

26. An audio signal processing apparatus according to claim 25, wherein said band-dividing filter means comprises a digital mirror filter.

FIG. 1

PRIOR ART

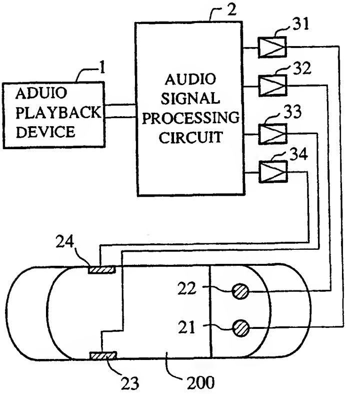

FIG. 2
PRIOR ART

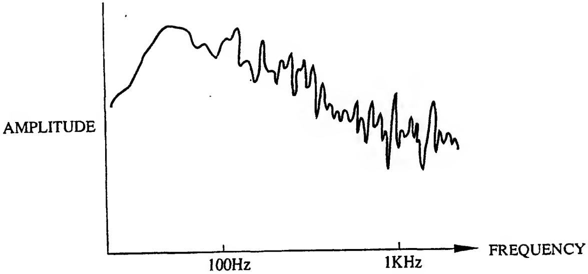

FIG. 3
PRIOR ART

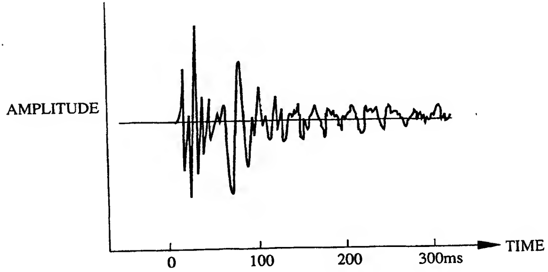

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FIG. 4
PRIOR ART

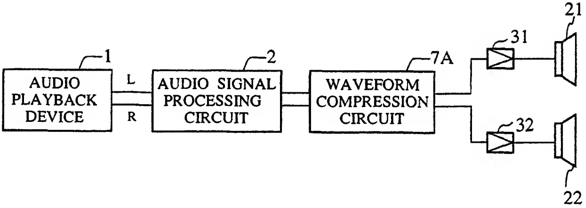


FIG. 5
PRIOR ART

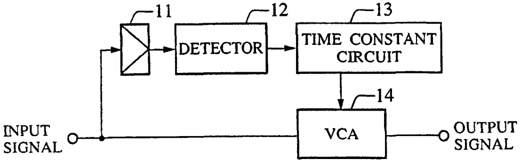
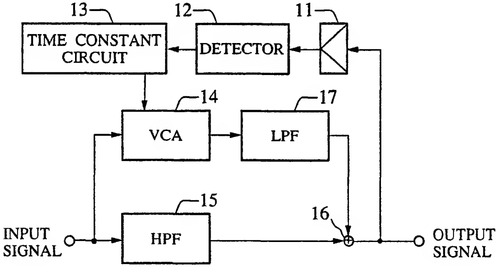


FIG. 6
PRIOR ART



6/29 2107320

FIG. 7

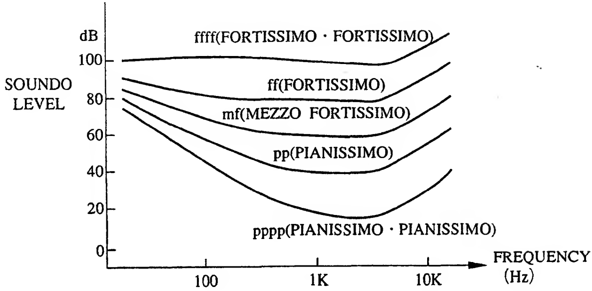
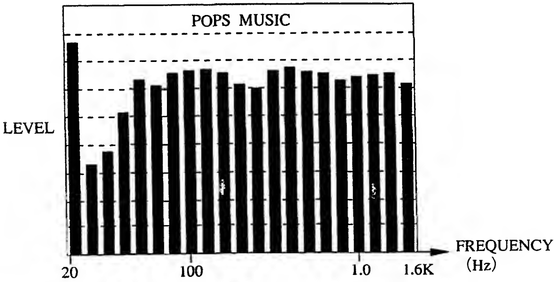


FIG. 8



Kirby, Eades, Gale, Baker

7|29 2107320

FIG. 9

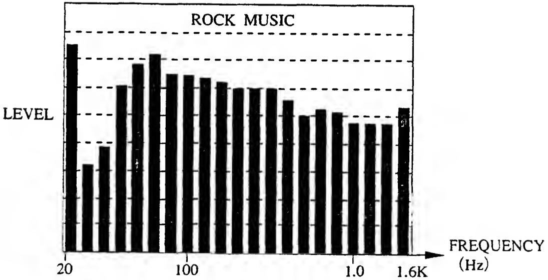


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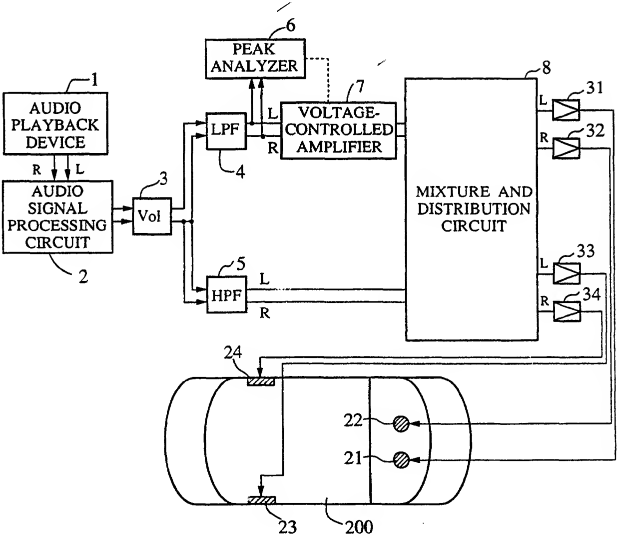
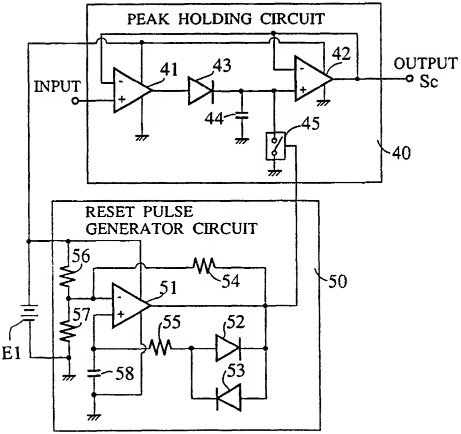
FIG. 10



6
PEAK ANALYZER
1
AUDIO PLAYBACK DEVICE
R
L
AUDIO SIGNAL PROCESSING CIRCUIT
2
3
Vol
LPF
4
L
R
7
VOLTAGE-CONTROLLED AMPLIFIER
8
MIXTURE AND DISTRIBUTION CIRCUIT
5
HPF
L
R
L
31
R
32
L
33
R
34
24
22
21
23
200

FIG. 11


PEAK HOLDING CIRCUIT
INPUT
41
43
42
OUTPUT
Sc
44
45
40
RESET PULSE
GENERATOR CIRCUIT
56
54
51
57
55
52
53
58
50
E1

10\29 2107320

FIG. 12

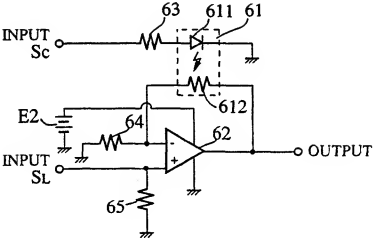


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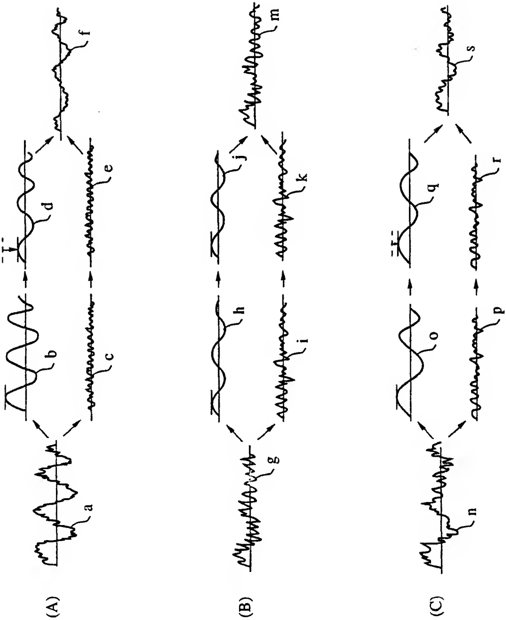
FIG. 13



12/29

2107320

FIG. 14

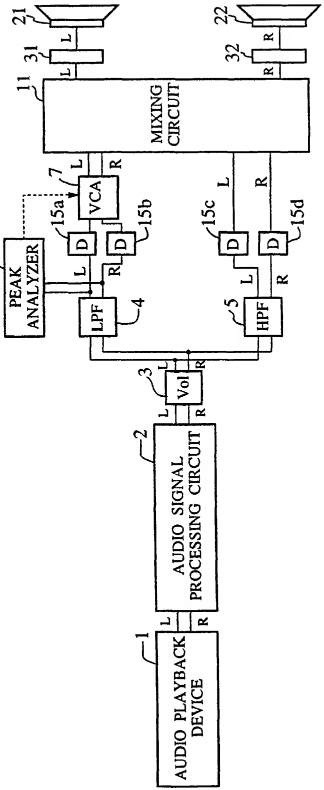
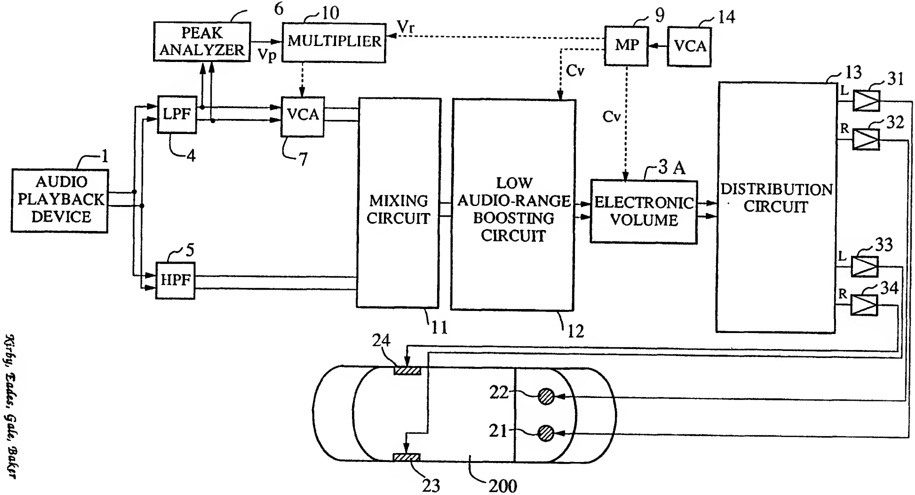


FIG. 15


AUDIO PLAYBACK DEVICE
1
LPF
4
HPF
5
PEAK ANALYZER
6
Vp
MULTIPLIER
10
Vr
VCA
7
MIXING CIRCUIT
11
LOW AUDIO-RANGE BOOSTING CIRCUIT
12
Cv
MP
9
VCA
14
Cv
ELECTRONIC VOLUME
3 A
DISTRIBUTION CIRCUIT
13
L
R
L
R
31
32
33
34
24
22
21
23
200

14/29

2107320

FIG. 16

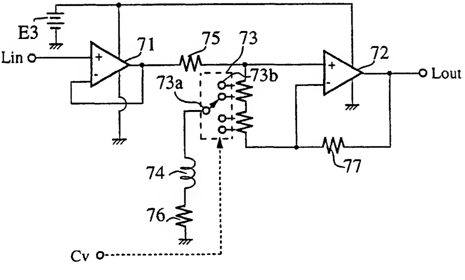


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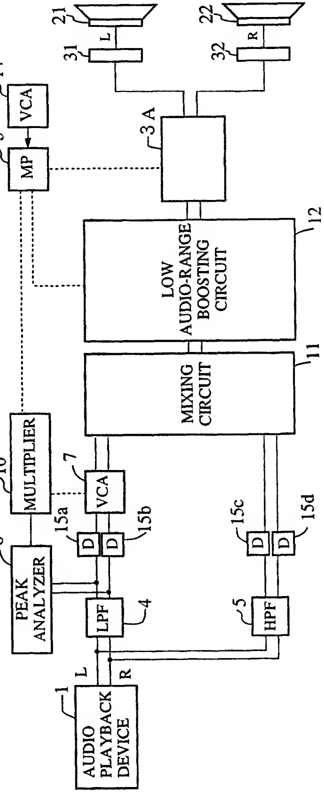
FIG. 17



AUDIO PLAYBACK DEVICE
1
L
R
LPF
4
HPF
5
PEAK ANALYZER
6
MULTIPLIER
10
VCA
7
D
15a
15b
15c
15d
MIXING CIRCUIT
11
LOW AUDIO-RANGE BOOSTING CIRCUIT
12
3 A
MP
9
VCA
14
31
32
21
22

FIG. 18

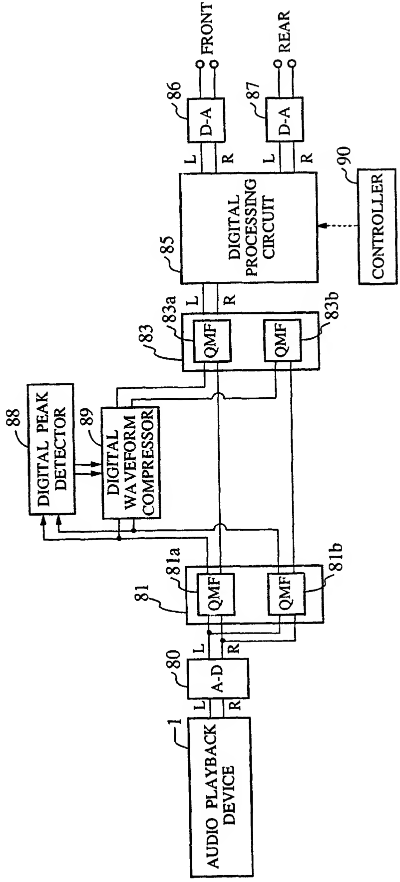

AUDIO PLAYBACK DEVICE
1
L
R
A-D
80
L
R
81
81a
81b
QMF
QMF
DIGITAL PEAK DETECTOR
88
DIGITAL WAVEFORM COMPRESSOR
89
83
83a
83b
QMF
QMF
L
R
DIGITAL PROCESSING CIRCUIT
85
CONTROLLER
90
L
R
L
R
D-A
86
D-A
87
FRONT
REAR

FIG. 19

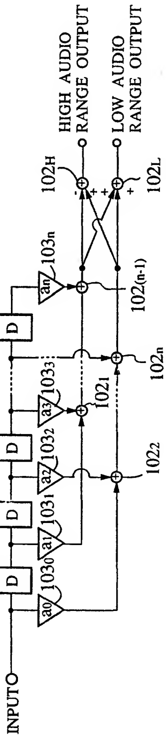
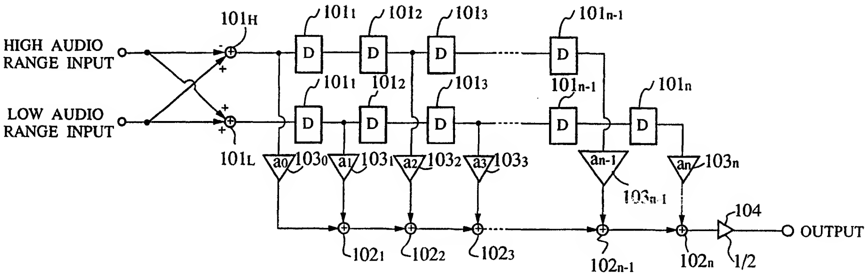

INPUT
D
D
D
D
101_1
101_2
101_3
101_n
a0
a1
a2
a3
an
103_0
103_1
103_2
103_3
103_n
102_1
102_2
102_n
$102_{(n-1)}$
102_H
102_L
HIGH AUDIO RANGE OUTPUT
LOW AUDIO RANGE OUTPUT

FIG. 20


HIGH AUDIO RANGE INPUT
LOW AUDIO RANGE INPUT
101H
101L
D
1011
1012
1013
101n-1
101n
a0
a1
a2
a3
an-1
an
1030
1031
1032
1033
103n-1
103n
1021
1022
1023
102n-1
102n
104
1/2
OUTPUT

2107320

19/29

FIG. 21

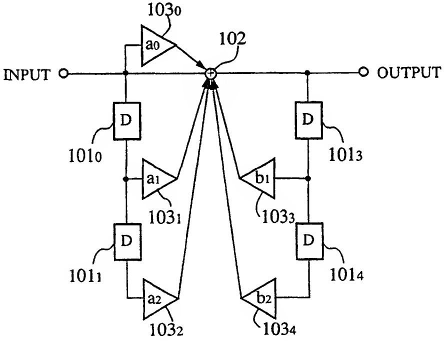


FIG. 22

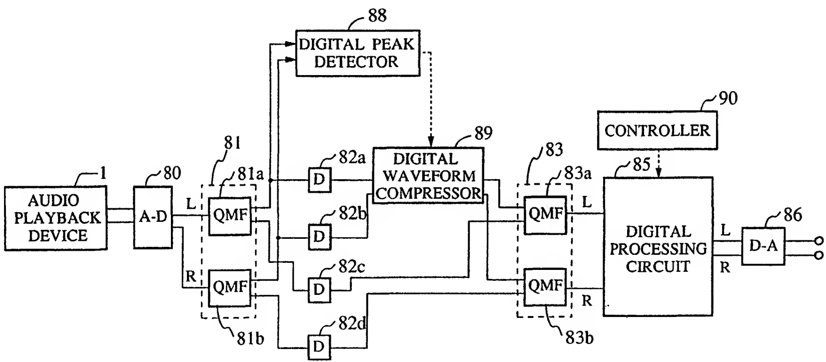
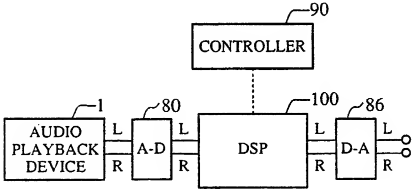

AUDIO PLAYBACK DEVICE
1
A-D
80
L
R
81
81a
81b
QMF
QMF
D
D
D
D
82a
82b
82c
82d
DIGITAL PEAK DETECTOR
88
DIGITAL WAVEFORM COMPRESSOR
89
83
83a
83b
QMF
QMF
L
R
CONTROLLER
90
85
DIGITAL PROCESSING CIRCUIT
L
R
D-A
86

FIG. 23


90
CONTROLLER
1
AUDIO PLAYBACK DEVICE
80
A-D
100
DSP
86
D-A
L
R
L
R
L
R
L
R

22/29 2107320

FIG. 24

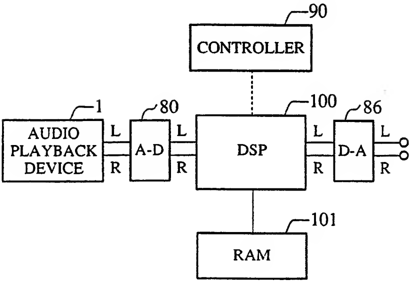


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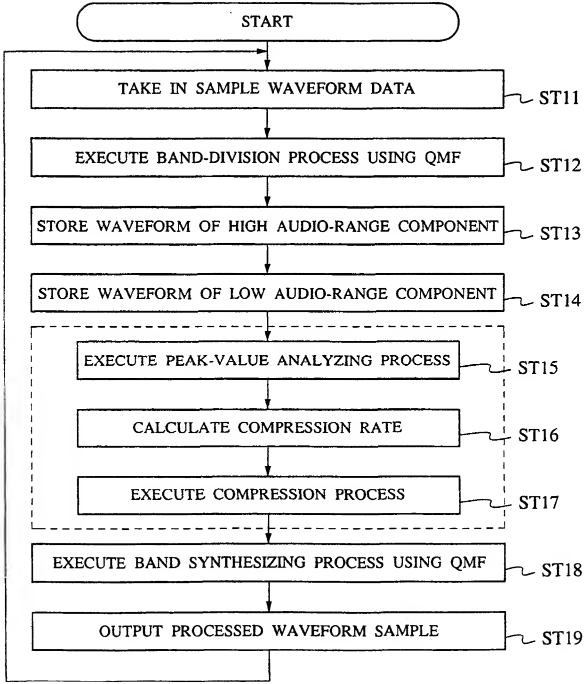
FIG. 25


START
TAKE IN SAMPLE WAVEFORM DATA
ST11
EXECUTE BAND-DIVISION PROCESS USING QMF
ST12
STORE WAVEFORM OF HIGH AUDIO-RANGE COMPONENT
ST13
STORE WAVEFORM OF LOW AUDIO-RANGE COMPONENT
ST14
EXECUTE PEAK-VALUE ANALYZING PROCESS
ST15
CALCULATE COMPRESSION RATE
ST16
EXECUTE COMPRESSION PROCESS
ST17
EXECUTE BAND SYNTHESIZING PROCESS USING QMF
ST18
OUTPUT PROCESSED WAVEFORM SAMPLE
ST19

24/29

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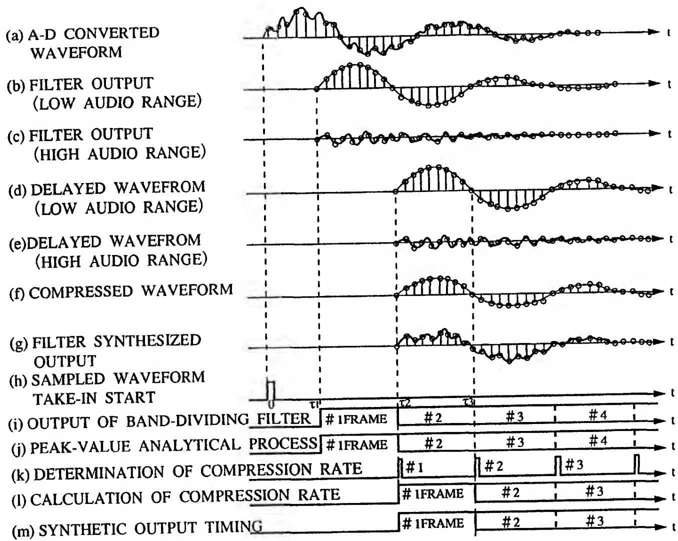
FIG. 26



Kirby, Eades, Gale, Baker

FIG. 27

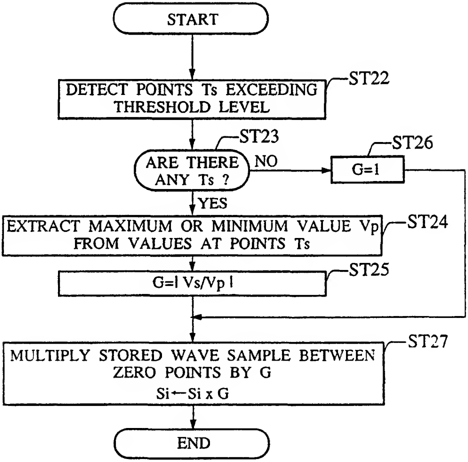
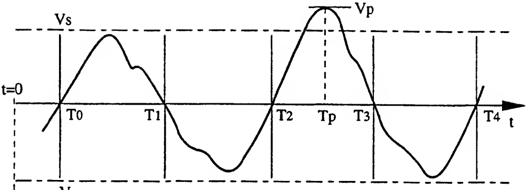
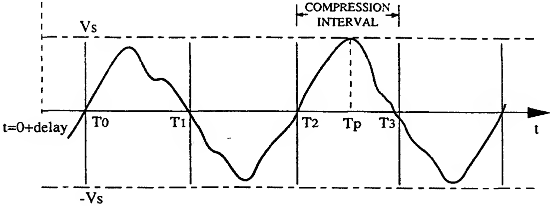

START
DETECT POINTS Ts EXCEEDING THRESHOLD LEVEL
ST22
ARE THERE ANY Ts ?
ST23
NO
G=1
ST26
YES
EXTRACT MAXIMUM OR MINIMUM VALUE Vp FROM VALUES AT POINTS Ts
ST24
G=| Vs/Vp |
ST25
MULTIPLY STORED WAVE SAMPLE BETWEEN ZERO POINTS BY G
Si←Si x G
ST27
END

FIG. 28

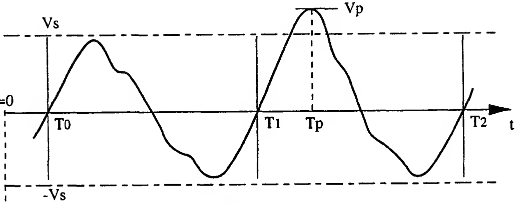

Vs
Vp
t=0
T0
T1
T2
Tp
T3
T4
t
-Vs

(a)

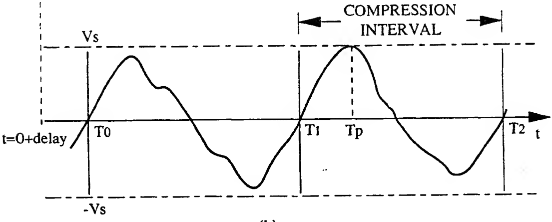

COMPRESSION
INTERVAL
Vs
t=0+delay
T0
T1
T2
Tp
T3
t
-Vs

(b)

FIG. 29


Vs
Vp
t=0
T0
T1
Tp
T2
t
-Vs

(a)


COMPRESSION
INTERVAL
Vs
t=0+delay
T0
T1
Tp
T2
t
-Vs

(b)

FIG. 30

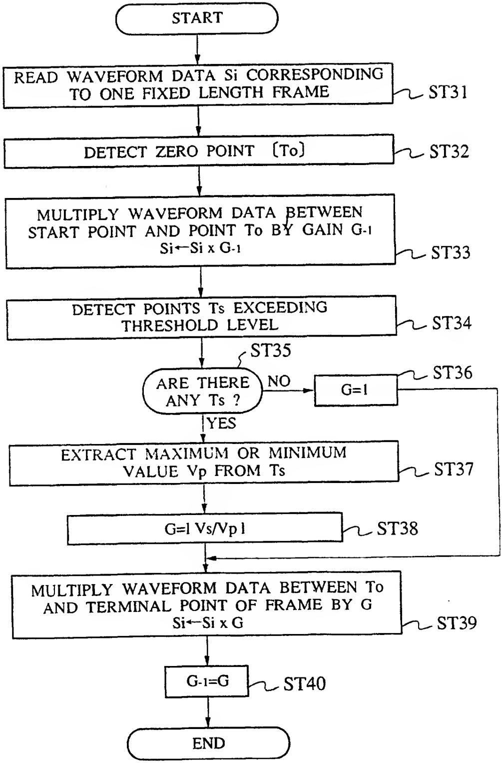
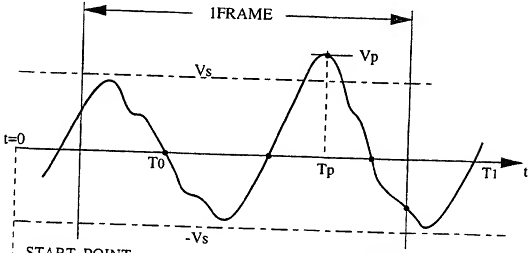
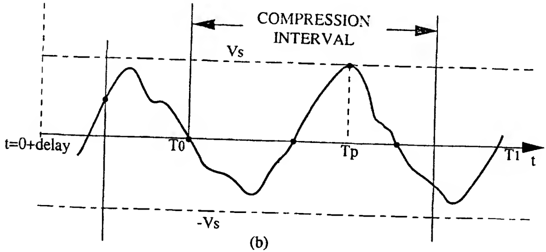

START
READ WAVEFORM DATA Si CORRESPONDING TO ONE FIXED LENGTH FRAME
ST31
DETECT ZERO POINT 〔To〕
ST32
MULTIPLY WAVEFORM DATA BETWEEN START POINT AND POINT To BY GAIN G-1
Si←Si x G-1
ST33
DETECT POINTS Ts EXCEEDING THRESHOLD LEVEL
ST34
ST35
ARE THERE ANY Ts ?
NO
G=1
ST36
YES
EXTRACT MAXIMUM OR MINIMUM VALUE Vp FROM Ts
ST37
G=| Vs/Vp |
ST38
MULTIPLY WAVEFORM DATA BETWEEN To AND TERMINAL POINT OF FRAME BY G
Si←Si x G
ST39
G-1=G
ST40
END

FIG. 31



1FRAME
Vp
Vs
t=0
T0
Tp
T1
t
-Vs
START POINT
TERMINAL POINT

(a)



COMPRESSION INTERVAL
Vs
t=0+delay
T0
Tp
T1
t
-Vs

(b)